Attornev's. Docket No. __99-885____

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Box Patent Application
Assistant Commissioner for Patents
Washington, D. C. 20231

NEW APPLICATION TRANSMITTAL

Transmitted herewith for filing is the patent application of:

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Man Li

For: METHOD AND SYSTEM FOR ESTIMATING PERFORMANCE
METRICS IN A PACKET-SWITCHED COMMUNICATION
NETWORK

Certification Under 37 CFR 1.10

I hereby certify that this New Application Transmittal and the documents referred to as enclosed therein are being deposited with the United States Postal Service on this date_____9/29/00____ in an envelope as "Express Mail Post Office to Addressee" mailing label EK673491939US addressed to the: Assistant Commissioner for Patents, Washington, D.C. 20231.

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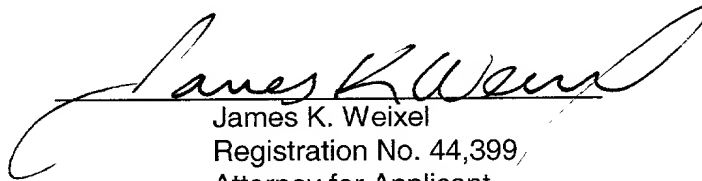
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Enclosed are:

[11]   pages of specification
[6]   pages of claims
[1]   page of abstract
[1]   sheet of formal drawings
[2]   pages of declaration and power of attorney.
[   ]   pages of assignment and assignment recordation form
[   ]   pages of information disclosure statement
[   ]   pages of form 1449
[   ]   references
[1]   return postcard

CLAIMS AS FILED				
	NUMBER FILED	NUMBER EXTRA	RATE	BASIC FEE $690.00
TOTAL CLAIMS	17 - 20	0 x	$18.00	0.00
INDEPENDENT CLAIMS	4 - 3	1 x	$78.00	78.00
MULTIPLE DEPENDENT CLAIMS(S)		0 x	$260.00	0.00
		TOTAL FILING FEE		$768.00

[x]   Please charge my Deposit Account No. 07-2339 in the amount of **$768.00. This transmittal letter is submitted in duplicate.**

[x]   The Commissioner is hereby authorized to charge any additional fees under 37 CFR 1.16 and 1.17 which may be required by the papers submitted herewith or credit any overpayment to Account No. 07-2339.
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METHOD AND SYSTEM FOR ESTIMATING PERFORMANCE METRICS IN A PACKET-SWITCHED COMMUNICATION NETWORK

REFERENCE TO RELATED APPLICATIONS

5   This application claims the benefit of U.S. Provisional Application No.

60/157,155, filed September 30, 1999, the entire contents of which are incorporated by

reference.

TECHNICAL FIELD

The present invention relates to packet-switched communication networks, and

10   more particularly to a method and system for estimating traffic between sources and

destinations in a communications network.

BACKGROUND OF THE INVENTION

Many Internet Service Providers ("ISPs") provide Enhanced Internet Protocol

15   ("EIP") real time services such as voice-over IP, fax-over IP, unified messaging, and

Internet call waiting over the IP network. The EIP real-time services differ from non-real

time, best-effort services provided over Transmission Control Protocol ("TCP"); unlike

best-effort services, EIP services have end-to-end quality of service (QOS) requirements

for packet loss, delay and delay jitter. As a result, providing EIP services requires new

20   processes for network resource planning.

Current processes are mostly based on a simple calculation of network link usage.

However, this method of planning may result in poor quality for EIP services because it

fails to address or estimate end-to-end QOS requirements, such packet loss, delay and delay jitter.

Although these problems have been addressed by prior art methods, many known methods make Poisson arrival and/or exponential packet length assumptions about

5   service traffic characteristics. In most cases, these assumptions are not applicable for EIP service traffic because different types of services have different packet length and inter-arrival distributions. Even for the same service, e.g. Voice over IP ("VoIP"), different voice encoders generate different traffic statistics. When silence compression capability is de-activated in the encoders, the resulting voice traffic consists of packets with

10   constant length and constant inter-arrival time. Once the silence compression capability is activated, packet inter-arrival times vary.

Given the mix of many types of deterministic and non-deterministic service traffic, approaches using large deviation theory to estimate end-to-end QOS have been suggested. However, large deviation theory-based estimates assume that there are a fixed

15   number of active sources feeding the network. For example, large deviation theory calculations may assume that there are n_1 voice calls, n_2 fax calls, and n_3 e-mail messaging calls, n_1, n_2, and n_3 being fixed numbers, which are active and that generate traffic at any given time. In reality, the number of active sources feeding the network varies over time because calls are constantly connected and disconnected over any given

20   time period, thereby increasing or decreasing, respectively, the total number of active calls. Thus, a direct application of the large deviation theory as taught by the known art is not the most appropriate method for estimating end-to-end QOS.

There is need for a process that estimates end-to-end packet loss, delay, and delay jitter for EIP network planning using the large deviation theory while taking into account changes in the number of active sources feeding the network.

5

SUMMARY OF THE INVENTION

Accordingly, the present invention is directed to a method for estimating end-to-end quality of service (QOS), such as packet loss, delay and delay jitter, in an enhanced IP network for use in network planning. The inventive method includes estimating the blocked traffic and carried traffic from each gateway in the network, determining a

10 possible number N of active sources, using a network routing algorithm to estimate the carried traffic for each network link, calculating at least one QOS characteristic for each network link by varying the number N for each calculation, and estimating an end-to-end QOS characteristic by summing the QOS characteristics for the network links. The inventive method therefore enhances network planning by explicitly taking into account

15 variances in the number of sources when estimating end-to-end QOS, thereby more reflecting real-world operation more accurately.

BRIEF DESCRIPTION OF THE DRAWING

Figure 1 is a flowchart illustrating one embodiment of the inventive method.

20

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Figure 1 illustrates a preferred embodiment of the inventive method. As is known in the art, an Internet Protocol ("IP") network contains inter-connected routers. To provide real time IP ("EIP") services, gateways are deployed at the edge of an IP

5 network. A gateway is connected to a public switched telephone network ("PSTN") through Primary Rate Interface ("PRI") links on one side and to the IP network on the other side. The gateway translates between the PSTN and IP networks.

During network planning, offered call rates to each gateway are estimated. Because each gateway has only a limited number of available PRI links, some of the

10 offered calls may be blocked. Moreover, calls of one service type may experience a higher blocking probability than calls of another type. An equation described in Kaufman, "Blocking in a Shared Resource Environment", IEEE Transactions on Communications, vol. COM-29, No. 10, October 1981, incorporated herein by reference, is used to determine the probability that calls will be blocked or carried by the network at

15 step 100. Kaufman's algorithm is preferred over an Erlang blocking model because Kaufman's algorithm gives blocking probabilities per type, whereas the Erlang model provides only the blocking probability for the aggregate of all calls.

The method generally involves first computing the QOS for each individual link and then using the individual link information to estimate the end-to-end QOS. More

20 particularly, steps 102 through 106 involve calculating QOS for each individual link, while steps 108 through 112 use the data obtained with respect to the individual links to estimate end-to-end QOS. At step 102, the process uses the routing algorithm of the IP

network to estimate, in Erlangs, the load carried by each link. As understood by those

skilled in the art, an Erlang is defined as the aggregate of continuous occupation of a

channel for one hour, where an intensity of one Erlang means the channel is continuously

occupied.

5   The packet loss at a router output link is calculated at step 104. In this step, it is

assumed that the bandwidth available to EIP services at a router link is C. In this

calculation, it is assumed that there are N_i independent sources of type i, where $i = 1$,

...,K. Two calls are of the same type if they present same traffic statistics, usually if they

employ the same type of voice encoder. For example, a VoIP call and a Voice messaging

10   call may be of the same type if they both use the ITU G.729 standard with silence

compression. For each source of type i, R_i and $\overline{R_i}$ denote the maximum and average

traffic rate of the type i source. Further, in the calculations below, $N = (N_1, ...N_k)$, $R_i =$

$(R_1, ...R_k)$, $\overline{R_i} = (\overline{R_1}, ... \overline{R_k})$. The value N can be either assumed or determined to take

on different values to take into account the fact that the number of active sources varies

15   over time in a real-world network.

Also in the calculation of packet loss for a single output link, for a single source

of type i, $A_i(t)$ denotes the number of arrivals of a single type i source in period (0,t). Its

asymptotic log moment generating function is:

20      $$\gamma_i(\Theta) = \lim_{t \to \infty} \frac{\log E[e^{\Theta A_i(t)}]}{t} \qquad (1)$$

The effective bandwidth function of the type i source is:

$$a_i(\Theta) = \frac{\gamma_i(\Theta)}{\Theta} \qquad (2)$$

5

Next, let $G = \{N : [N \bullet R] > C \text{ and } [N \bullet \overline{R}] < C\}$, where $[N \bullet R]$ denotes a scalar product of N and R. If the scalar product $N \bullet R$ is less than C, there is no queuing, and therefore no packet delay even if all of the sources transmit at peak rate because the queue has not yet been saturated; in this case, packet delay is caused only by transmission delay. If the scalar product is greater than C, however, the queue is saturated and will experience delays due to an overload of the link. Next, the following equation is solved for Θ. $\delta(N)$ is defined as the solution for this equation and will be referenced below in equation (4).

15

$$\sum_{i=1}^{K} N_i a_i(\Theta) = C \qquad N \in G \qquad (3)$$

20

Next, b is defined as the buffer size of the router output buffer assigned to EIP services. According to large deviation theory, which is known in the art:

$$P(Q > b|N) \sim e^{-b\delta(N)} \qquad \text{as } b \longrightarrow \infty \qquad (4)$$

25

where $P(Q>b|N)$ is the loss probability conditioned on N.

The above definitions and calculations will now be used to calculate the packet

loss at the router output link, taking into account changes in the number of active sources

5   N over time. More particularly, when a type i call is admitted into the network and starts

to generate packets, it becomes a new active source of type i. When the call completes,

the number of active sources N is reduced by one; as a result, N varies over time as calls

enter and leave the network. During peak times, there are typically a large number of

active sources feeding the link. A Poisson distribution can model the number of active

10   sources in the link as follows:

$$P(N) = \prod_{i=1}^{K} \frac{\rho_i^{N_i} e^{-\rho_i}}{N_i!} \qquad (5)$$

15

where ρ_i is the offered load of type i service in Erlang. Removing the dependence on N,

the probability of packet loss at a router output link is as follows:

20    $$P(Q > b) = \sum_{N \in G} P(N) \times e^{-b\delta(N)} \quad + \quad \sum_{N \in \{[N \cdot R]\} > C} P(N) \qquad (6)$$

25   The calculation in equation (6) takes into account the fact that N varies over time by

summing, over all possible values of N, the less probabilities $P(Q>b|N)$ weighted by the

probability that N occurs. This expression also assumes that the number of active sources

will vary slower than the contents in the buffer. As a result, this estimate more closely

reflects the way in which active sources feed into the link.

Next, at step 106, the inventive method calculates the packet delay for an

individual router output link. In this calculation, D is a random variable representing the

packet delay at a router output link. The packet delay distribution may be obtained from

packet loss at the router output link, which was calculated in step 104, as follows:

5

$$P(D>d) = P(Q/C>d) = P(Q>dC) \tag{7}$$

where C is the output link speed.

At this point, the probability of both packet loss and packet delay at an individual

10   router output link has been calculated for all possible values of N active sources, thereby

taking into account changes in the number of active sources in the network. Next, steps

108 through 112 take the single link statistic calculations from steps 102 through 106 and

use them to calculate end-to-end packet statistics.

Step 108 of the inventive method involves calculating the end-to-end packet loss.

15   In this calculation, it is assumed that the losses occur independently along the end-to-end

path and that there are a total number of L router output links. Preferably, the loss and

delay characteristics for each one of the L router output links is calculated according to

steps 104 and 106 described above. With respect to the end-to-end packet loss

calculation at step 108, p_i is defined as the loss probability of an ith link along the path.

20   The total end-to-end loss probability along the path is then calculated to be:

$$1 - \prod_{i=1}^{L} (1 - p_i) \tag{8}$$

25

Next, the end-to-end packet delay is calculated at step 110. In this calculation, it is assumed that the delay for each individual packet occurs independently, without being influenced by delays in other packets along the path. To calculate the end-to-end packet delay, $f_i(x)$ is the delay distribution of the ith link along the path. The end-to-end delay distribution along the path is therefore defined as:

$$f(x) = f_1(x) \otimes f_2(x) \ldots \otimes f_L(x) \qquad (9)$$

where f(x) is the end-to-end delay distribution and $\otimes$ denotes convolution. More specifically, the end-to-end delay distribution is the convolution of the link delay distributions along the path.

Next, at step 112, the inventive process calculates the end-to-end packet delay jitter. In this step, the user can select a specific delay jitter percentage p for the computation (e.g., 95%, 98% or 99% delay jitters). Delay jitter is defined as t_2-t_1, where $(100-p)/2$ percent of packets experience delays less than t_1 ms and $(100-p)/2$ percent of packets experience delays greater than t_2 ms. In particular, t_1 and t_2 are found by solving the following equations.

$$\int_0^{t_1} f(x)dx = p/2 \qquad\qquad \int_{t_2}^{\infty} f(x)dx = p/2 \qquad (10)$$

Thus, the inventive method estimates end-to-end QOS by first estimating the amount of traffic blocked and carried by the network, preferably using Kaufman's algorithm, estimating single link statistics for each individual router output link in a path,

taking into account variations in the number of active sources, and then using the single

link statistics to calculate end-to-end QOS. In these calculations, it is assumed that the

estimates of offered traffic for each service to each gateway is given, preferably in

Erlangs. By calculating the loss and delay probability for all possible N number of active

5   sources, as described by way of example in Equations (5) through (10), the inventive

method provides a better estimate of network traffic because it considers the fact that the

number of active sources will vary over time.

The inventive method can be carried out in any known network system. The

system can include a database that contains the parameters (e.g. service type and

10   characteristics) for each gateway in the system, a memory that stores an end-to-end

quality service program corresponding to the inventive method, and a processor that

executes the end-to-end quality service program according to the steps shown in Figure 1.

From the inventive method, the system can output network information such as Wide

Area Network ("WAN") link utilization; one-way packet loss; one-way and two-way

15   packet delay and delay jitter; gateway Primary Rate Interface ("PRI") link utilization and

blocking probabilities per physical location; and/or global maximum PRI link utilization

and maximum blocking probability.

An Internet Service Provider ("ISP") can therefore estimate network performance

using the inventive method and system and, if QOS requirements are not met, reallocate

20   sources and conduct the inventive method again quickly and easily to assess the effect of

the reallocation on network performance. For example, the ISP may experiment with

different traffic scenarios to distribute global origination traffic among network cities in different ways.

It should be understood that various alternatives to the embodiments of the invention described herein may be employed in practicing the invention. It is intended that the following claims define the scope of the invention and that the method and apparatus within the scope of these claims and their equivalents be covered thereby.

CLAIMS

What is claimed is:

1.   A method for estimating end-to-end quality of service parameters for a

Internet Protocol ("IP") communications network, wherein the end-to-end quality of

service parameters are selected from end-to-end packet loss, end-to-end packet delay and

end-to-end packet delay jitter, and wherein the IP communications network includes

5   routers connected by links and gateways between a Public Switched Telephone Network

("PSTN") and the routers, the method using estimates of offered traffic for each service

from each source carried to each gateway of the IP communications network, comprising

the steps of:

determining a possible number N of sources;

10   estimating, from the offered traffic to each gateway, blocked traffic and carried

traffic from each gateway;

estimating carried traffic for each network link using a network routing algorithm;

calculating a plurality of loss probabilities by varying the number N for each

calculation;

15   estimating an end-to-end packet loss probability by summing the plurality of loss

probabilities from the calculating step.

2.   The method of claim 1, wherein the plurality of loss probabilities is

calculated over all possible values of N, and wherein the step of estimating an end-to-end

packet loss probability sums the plurality of loss probabilities over all possible values of N.

3.   The method of claim 1 further comprising the step of estimating the single link packet delay distribution after the step of estimating the carried traffic for each network link.

4.   The method of claim 1 further comprising the step of estimating the end-to-end packet delay distribution.

5.   The method of claim 1 further comprising the step of estimating the end-to-end packet delay jitter.

6.   The method of claim 1 wherein the offered traffic for each service has deterministic and non-deterministic attributes for packet length and inter-arrival distribution.

7.   The method of claim 1 wherein a Kaufman algorithm is used in the step of estimating blocked traffic and carried traffic.

8.   The method of claim 1 wherein the step of estimating end-to-end packet loss probability includes estimating a single link packet loss probability for each link.

9.   The method of claim 1, wherein the estimates of offered traffic are in Erlangs.

10.   A method for estimating end-to-end quality of service parameters for a Internet Protocol ("IP") communications network, wherein the end-to-end quality of service parameters are selected from end-to-end packet loss, end-to-end packet delay and end-to-end packet delay jitter, and wherein the IP communications network includes

5   routers connected by links and gateways between a Public Switched Telephone Network ("PSTN") and the interconnected routers, the method using estimates of offered traffic having deterministic and non-deterministic attributes for packet length and inter-arrival distribution for each service from each of a varying number of sources carried to each gateway of the IP communications network, comprising the steps of:

10   determining a possibly number N of sources;

estimating, from the offered traffic to each gateway, a blocked traffic and a carried traffic from each gateway;

estimating carried traffic for each network link using a network routing algorithm;

estimating a single link packet loss probability for each link;

15   calculating a plurality of loss probabilities by varying the number N for each calculation;

estimating an end-to-end packet loss probability by summing the plurality of loss probabilities;

estimating the single link packet delay distribution;

20   estimating the end-to-end packet delay distribution; and

estimating the end-to-end packet delay jitter.

11.   The method of claim 10, wherein the plurality of loss probabilities is

calculated over all possible values of N, and wherein the step of estimating an end-to-end

packet loss probability sums the plurality of loss probabilities over all possible values of

N.

12.   A method for estimating end-to-end quality of service parameters for a

Internet Protocol ("IP") communications network, wherein the end-to-end quality of

service parameters are selected from end-to-end packet loss, end-to-end packet delay and

end-to-end packet delay jitter, and wherein the IP communications network includes

5   routers connected by links and gateways between a Public Switched Telephone Network

("PSTN") and the routers, the method, using estimates of offered traffic for each service

from each source carried to each gateway of the IP communications network, comprising

the steps of:

determining a possible number N of sources;

10   estimating, from the offered traffic to each gateway, a blocked traffic and a

carried traffic from each gateway;

estimating carried traffic for each network link using a network routing algorithm;

calculating at least one quality of service parameter for each network link by

varying the number N for each calculation; and

15   estimating at least one end-to-end quality of service parameter by summing the

quality of service characteristics for the network links.

13.   The method of claim 12, wherein the quality of service characteristic for

the network link is selected from the group consisting of single link packet loss

probability and single link packet delay distribution.

14.   The method of claim 12, wherein the end-to-end quality of service

characteristic is selected from the group consisting of end-to-end packet loss probability,

end-to-end packet delay distribution, and end-to-end packet delay jitter.

15.   The method of claim 14, wherein the quality of service characteristic for

the network link is selected from the group consisting of single link packet loss

probability and single link packet delay distribution.

16.   The method of claim 12, wherein the estimates of offered traffic are in

Erlangs.

17.   A system for estimating end-to-end quality of service parameters for a

Internet Protocol ("IP") communications network, wherein the end-to-end quality of

service parameters are selected from end-to-end packet loss, end-to-end packet delay and

end-to-end packet delay jitter, and wherein the IP communications network includes

5   routers connected by links and gateways between a Public Switched Telephone Network

("PSTN") and the routers, the system, using estimates in Erlang, of offered traffic for

each service from each source carried to each gateway of the Internet Protocol

communications network, comprising:

    a database comprising parameters for each gateway, the parameters identifying

10   type of service and characteristics of each service;

    a memory comprising an end-to-end quality of service program; and

    a processor utilizing the end-to-end quality of service program to

        determine a possible number N of sources;

        estimate, from the offered traffic to each gateway, a blocked traffic and a

15   carried traffic from each gateway;

        estimate carried traffic for each network link using a network routing

algorithm;

        calculate a plurality of loss probabilities by varying the number N for each

calculation; and

20   estimate an end-to-end packet loss probability by summing the plurality of

loss probabilities.

ABSTRACT

A method for estimating end-to-end quality of service (QOS), such as packet loss, delay, and delay jitter, in a packet-switched communications network includes steps for calculating packet loss and packet delay each router output link in a network path and

5   using the packet loss and packet delay calculations for each individual router output link to estimate end-to-end QOS. The method includes calculating the loss probability for all possible numbers of active sources to take into account changes in the number of active sources as active sources connect to and disconnect from the network. As a result, the inventive method allows the number of active sources to vary in its estimation of end-to-

10   end packet loss, delay, and delay jitter for enhanced Internet Protocol network planning.
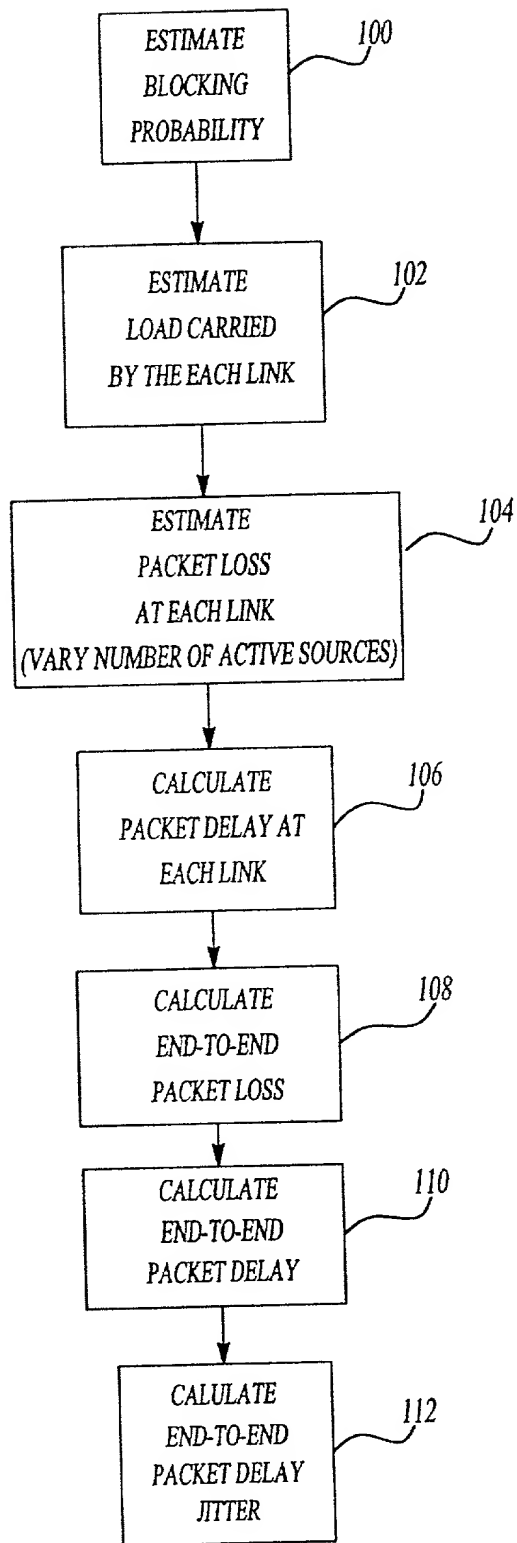
ESTIMATE
BLOCKING
PROBABILITY — 100

↓

ESTIMATE
LOAD CARRIED
BY THE EACH LINK — 102

↓

ESTIMATE
PACKET LOSS
AT EACH LINK
(VARY NUMBER OF ACTIVE SOURCES) — 104

↓

CALCULATE
PACKET DELAY AT
EACH LINK — 106

↓

CALCULATE
END-TO-END
PACKET LOSS — 108

↓

CALCULATE
END-TO-END
PACKET DELAY — 110

↓

CALULATE
END-TO-END
PACKET DELAY
JITTER — 112

Fig-1

DECLARATION AND POWER OF ATTORNEY FOR PATENT APPLICATION

Attorney Docket No. **99-885**

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name,

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled:

METHOD AND SYSTEM FOR ESTIMATING PERFORMANCE METRICS IN A PACKET-SWITCHED COMMUNICATION NETWORK

the specification of which (check one) [X] is attached hereto. [ ] was filed on
as Appln. Serial No.   and was amended on
(if applicable). I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the patentability of this application in accordance with Title 37, Code of Federal Regulations, Section 1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, Section 119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

Prior Foreign Application(s)   Priority Claimed
_____ _____ _____ [_] Yes [_] No
(Number)   (Country)   (Day/Month/Year filed)

I hereby claim the benefit under Title 35, United States Code, 119(e) of any United States provisional applications(s) listed below.

60/157,155   9/30/99
(Application Number)   (Filing Date)

I hereby claim the benefit under Title 35, United States Code, Section 120 of any United States application(s) listed below and insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, Section 112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, Section 1.56 which occurred between the filing date of the prior application and the national or PCT international filing date for this application:

_____ _____ _____
(Appln. Serial No.)   (Filing Date)   (Status--patented, pending, abandoned)

I hereby appoint the following attorney(s) and/or agent(s) to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith:

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I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

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